OIPE JC99
JUN 1 1 2001
PATENT & TRADEMARK OFFICE

PATENT APPLICATION

#4

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re the Application of

Roman VITENBERG

Application No.: 09/839,205

Filed: April 23, 2001 Docket No.: 109335

For: COMMUNICATION SYSTEMS CONVEYING VOICE AND DATA SIGNALS OVER STANDARD TELEPHONE LINES

CLAIM FOR PRIORITY

Director of the U.S. Patent and Trademark Office
Washington, D.C. 20231

Sir:

The benefit of the filing date of the following prior foreign application filed in the following foreign country is hereby requested for the above-identified patent application and the priority provided in 35 U.S.C. §119 is hereby claimed:

Israel Patent Application No. 135794 filed April 23, 2000

In support of this claim, a certified copy of said original foreign application:

___X___ is filed herewith.

_______ was filed on _____ in Parent Application No. _____ filed _____.

_______ will be filed at a later date.

It is requested that the file of this application be marked to indicate that the requirements of 35 U.S.C. §119 have been fulfilled and that the Patent and Trademark Office kindly acknowledge receipt of this document.

Respectfully submitted,

James A. Oliff
Registration No. 27,075

Thomas J. Pardini
Registration No. 30,411

JAO:TJP/cmm
Date: June 11, 2001

OLIFF & BERRIDGE, PLC
P.O. Box 19928
Alexandria, Virginia 22320
Telephone: (703) 836-6400

DEPOSIT ACCOUNT USE AUTHORIZATION
Please grant any extension necessary for entry;
Charge any fee due to our Deposit Account No. 15-0461

OIPE JC96
JUN 11 2001
PATENT & TRADEMARK OFFICE

מדינת ישראל
STATE OF ISRAEL

Ministry of Justice
Patent Office

משרד המשפטים
לשכת הפטנטים

This is to [illegible] that annexed [illegible]eto is a [illegible]opy of the [illegible]ments [illegible] originally [illegible]ited wi[illegible] the patent application of which particulars are specified on the first page of the annex.

זאת לתעודה כי רצופים בזה העתקים נכונים של המסמכים שהופקדו לכתחילה עם הבקשה לפטנט לפי הפרטים הרשומים בעמוד הראשון של הנספח.

This 1 0 -05- 2001 היום

רשם הפטנטים
Commissioner of Patents

נתאשר
Certified

חוק הפטנטים, התשכ"ז–1967
PATENTS LAW, 5727-1967

בקשה לפטנט
Application for Patent

לשימוש הלשכה For Office Use	
מספר: Number	135791
תאריך: Date	23-04-2000
הוקדם/נדחה Ante/Post-dated	

אני, (שם המבקש, מענו – ולגבי גוף מאוגד – מקום התאגדותו)
I (Name and address of applicant, and, in case of body corporate-place of incorporation)

בעל אמצאה מכח ROMAN VITENBERG ששמה הוא
Owner, by virtue of ... of an invention, the title of which is

(בעברית)
(Hebrew)

(באנגלית)
(English)

"A METHOD AND APPARATUS FOR TRANSMITTING OF VOICE AND DATA OVER SUBSCRIBER TWISTED PAIR."

מבקש בזאת כי ינתן לי עליה פטנט
hereby apply for a patent to be granted to me in respect thereof.

• בקשת חלוקה – Application of Division	• בקשת פטנט מוסף – Application for Patent Addition	• דרישה דין קדימה Priority Claim		
מבקשת פטנט from Application No ______ מס' dated ______ מיום	• לבקשה/לפטנט to Patent/Appl. No ______ מס' dated ______ מיום	מספר/סימן Number/Mark	תאריך Date	מדינת האגוד Convention Country
• יפוי כח: כללי/מיוחד – רצוף בזה / עוד יוגש P.O.A.: general / individual – attached / to be filed later – filed in case ______ הוגש בענין				
המען למסירת הודעות ומסמכים בישראל Address for Service in Israel ROMAN VITENBERG HADAR ST. 9 APT 6 HOLON 58210 ISRAEL.				

התימת המבקש
Signature of Applicant

היום 16 בחודש APRIL שנת 2000
This 16 of APRIL of the year 2000

לשימוש הלשכה
For Office Use

A METHOD AND APPARATUS FOR TRANSMITTING OF VOICE AND DATA OVER SUBSCRIBER TWISTED PAIR.

ABSTRACT

A communication system uses DMT line signals for simultaniosly transmitting a data and telephone signal over subscriber twisted pair. A data and voice are transmitted on different carriers of DMT line signal. In during initialization process a communication system assignes for voice transmitting part of DMT carriers those have high signal to noise ratio and are able to carry more than n-bit per QAM symbol. A voice is transformed in digital form by PCM coder that produce n-bit PCM words. Each n-bit PCM word is transmitted by n-bit QAM symbol on carriers assigned to the voice transmission. A convertion of n-bit PCM word to the n-bit QAM symbol is provided in such way that probability of error is minimal for more significant bits of said PCM word. In basic embodiment a communication system comprises first tone ordering block and first constellation encoder to modulate by voice a number of carriers of DMT assigned to voice transmitting and second tone ordering block and second constellation encoder to modulate others carriers of DMT by data.

A communication system in one embodiment of the invention comprises carriers allocation block that connected to first tone ordering block and to second tone ordering block to provide dynamic allocation of data on said carriers assignated for voice transmitting in during time intervals those are not used for telephone communication.

A communication system in another embodiment comprises PCM interface port that connected directly to PCM telephone equipment of CO.

FIELD OF THE INVENTION

The present invention relates generally to voice and data communications over subscriber twisted pair of telephone cable and, more particularly, to a method and apparatus for voice communication over asymmetrical digital subscriber line (ADSL).

BACKGROUND OF THE INVENTION

Asymmetric Digital Subscriber Line (ADSL) is a new communication technology that allows existing twisted pair Cable Telephone Network to be converted into a high-performance Multimedia Digital Network for multimedia and high-speed data communications with ability provide to every subscriber high speed data communication that include many new services as Video on demand, Conference Video-Phone, HDTV Broadcast, Digital Hi Fi Audio, Fast Internet and e.g.

ADSL is point to point connected circuit and includes an ADSL modem on each end of the twisted pair telephone line, creating three information channels - a high speed downstream (central office to end user) channel, a medium speed upstream (end user to central office) channel, and a POTS ("Plain Old Telephone Service") channel. The POTS channel is separated from the ADSL modem by filters, thus guaranteeing uninterrupted POTS, even if the ADSL circuit fails. While description is provided in terms of the POTS channel, telephone voice communications signals, telephone instruments, and the like for the benefit of familiarity, it should be understood that telephone equipment and signals need not be limited to voice communications, but may also include other technologies, for example equipment and signals compatible with regular telephone lines, such as facsimiles machines,

voice band modems (for example, V.90 modems), answering machines, and the like.

Two variants of ADSL systems are available today - full-rate ADSL in accordance with the T1E1.413 or ITU G.992.1 standards and "splitterless" ADSL defined by the ITU G.992.2 standard. Full-rate ADSL uses POTS splitters to separate the POTS channel from the ADSL data signals. A POTS splitter is installed at each end of the line and includes a lowpass filter for separating out POTS telephone voice communication signals and a high pass filter for separating out data communication signals.

The POTS splitter divides the subscriber line into two separate twisted pairs - one for data communication (ADSL) and one for telephone voice communication signals (POTS). As a result, the existing two-wire internal house telephone wiring is not usable for ADSL. New wiring must be installed from the splitter to the modem, resulting in increased installation cost.

Splitterless ADSL can be installed without the need for additional home wiring. In this case, the ADSL modem includes a high-pass filter that rejects the POTS telephone voice communication signal, while every telephone instrument in the house is connected to the telephone line through a low-pass microfilter that rejects the ADSL line signals.

FIG. 1 is a block diagram illustrating a splitterless ADSL system **101** of the prior art. A subscriber premise **103** is coupled to central office (CO) **109** by twisted pair **107** of telephone cable. At the subscriber premise **103** said twisted pair **107** is connected to a fax **121**, to a telephone **123** and to a remote ADSL Transceiver Unit (ATU-R) **105** using internal telephone lines **117**. ATU-R **105** is connected directly to the telephone line **117** and to personal computer (PC) **125** by Ethernet cable **124**. A fax **221** and a telephone **123** are connected to telephone line **117** by microfilters **219**. A Central Office (CO) **109** includes a central office ADSL Transceiver Unit (ATU-C) **111**, a POTS-splitter **131**, POTS Line Card **108**, data switch **135**, telephone switch **137**, data network **115**, and telephone network **113**. A subscriber twisted pair **107** is coupled to POTS-splitter **131** that separates data and voice signals. A data signal passes to ATU-C **111** that coupled to the data switch **135**. A voice signal passes to POTS Line Card **108** that coupled to telephone switch **137**. The telephone switch **137** is coupled to a telephone network **113** and a data switch **135** is coupled to data network **115**.

Voice communications passing through telephone switch **137** are passed through POTS splitter **131** and applied to twisted pair **107** as baseband signals. Data communications passing through data switch **135** are modulated at a frequency range higher than that of the baseband POTS signals and passed through POTS splitter **131** and applied to twisted pair **107**. Since the data communications are transmitted at a different frequency range than the voice communications, frequency-division-multiplexing (FDM) allows simultaneous transmission of both voice communications (POTS) and data communications over a single twisted pair **107**.

Fig.2 is a block diagram of an ADSL Transceiver Unit (ATU) transmitter showing the functional blocks and interfaces that are referenced in ITU Recommendation G.992.2.

An information data **153** comes to a data interface port **151** that produce a sequence of ATM cells **152**. Said cells come to the coder block **155** that produces scrambling and Reed Solomon coding of ATM cells. After that a coded data stream **156** comes to interleaver **157** that mixes data bits to protect an information data from impulse noise. An interleaved data stream **158** passes to tone ordering block **159** that distributes this data on **128** tones (carriers) of multitone line signal. A constellation encoder and a gain scaling block **161** calculates modulation parameters for each carrier in accordance with data that must be transmitted by said carrier. Inverse Fourier Transformer block **163** transforms said modulation parameters of all carriers to digital samples of DMT signal and writes said digital samples into output buffer **165**. A digital to analog converter **167** transforms said digital samples to analog DMT line signal.

As well known ADSL is an adaptive system. In during initialization phase of communication an ADSL system measures a signal to noise ratio for each carrier and define number of bit that may be loaded on the carrier.

Fig.3 illustrates typical graph of ADSL downstream and upstream bit-loading. Practically for cable length up to 9000 fits many carriers may be loaded with high number of bit per symbol (up to 12 – 14 bit).

A standard ADSL system has some disadvantages and problems.

One problem of existing ADSL system is that for simultaneously transmit of data and voice is necessary to use POTS-splitter and POTS Line Card those are expensive devices.

In during last time was developed an " Integrated ADSL Line Card " that realizes a digital processing of ADSL signal and of a voice signal on the same board. In this case a

separation of ADSL line signal and a voice signal produces without POTS-splitter by digital signal processor.

Said method was described in USA Patent US5589856 " ADSL integrated line card with digital splitter and POTS CODEC without bulky analog splitter".

Said "Integrated ADSL Line Card " may decreases cost of ADSL equipment but is very difficult for realization because such device must transmit a composite signal of both the high-frequency ADSL data and voice by the same driver circuits.

Another problem of an existing ADSL system is that only one baseband voice channel may be provided to customer. The reason is that ADSL was developed for data transmission and is wrong adapted for voice transmitting. A splitterless ADSL in accordance with the ITU G.992.2 standard comprises interleaving device that inserts big delays into transmitted data. A voice transmitting in a digital form over ADSL system in this case is difficult.

SUMMARY OF THE INVENTION

The present invention is directed to a method and apparatus for data and voice communication in an asymmetric digital subscriber line (ADSL) system. In accordance with the present invention, the disadvantages and problems associated with voice transmitting over ADSL system have been substantially reduced or eliminated. In particular, several high quality telephone channels may be inserted into ADSL system without necessary to change an existing standard.

In accordance with present invention a voice and data signals are simultaneously transmitted on different carriers of DMT ADSL line signal.

A voice channel transmits by two carriers of downstream and two carriers of upstream that may carry equal or more than 8-bit per QAM symbol. In during of training process, ADSL system analyzes signal to noise ratio (SNR) in each carrier and assignees two downstream and two upstream carriers with SNR > 44 dB for voice transmission. Such carriers in following names "voice carriers". Others carriers of the DMT line signal are assigned for data transmitting ("data carriers"). After initialization ADSL system uses "data carriers" for transmitting of data in accordance with existing standard ITU G.992.1 or ITU G.992.2.

Before transmitting a voice signal is converted by PCM coder into 8-bit digital words in accordance with existing standard for PCM telephone system (T1 or E1). A PCM coder works synchrony with ADSL frame and produce 8-bit words with frequency 8 kHz. Each 8-bit PCM word is converted into 8-bit QAM symbol. Odd QAM symbols modulate first "voice carrier", even QAM symbols modulate second "voice carrier". A conversion of 8-bit PCM word into 8-bit QAM symbol is produced in such way that probability of error is minimal for more significant bits of said PCM word.

It is clear that several voice channels may be transmitted by same way. For short distance (9kfit) a standard ADSL system may support simultaneously up to 8 telephone channels, data upstream with bit rate up to 250 kb/s and data downstream with bit rate up to 6Mb/s.

A communication system in accordance with basic embodiment of the present invention comprises a home ADSL Transceivers Unit (ATU-R) placed at the subscriber premise and connected by twisted pair of telephone cable to an office ADSL Transceivers

Unit (ATU-C) placed at the Central Office of telephone station. ATU-R comprises ADSL transmitter, PCM coder and synchronization circuit to provide synchronization between frames of ADSL DMT signal and sampling rate of PCM coder. The ADSL transmitter includes circuits for separate loading data and voice channels on "data carriers" and "voice carriers" of DMT line signal. The ADSL transmitter comprises a device for modulating of "voice carriers" with constellation table that minimize a noise in the voice channel. ATU-R comprises data interface port to provide data service to customer and voice interface port to provide telephone service to customer.

ATU-C comprises ADSL transmitter and PCM coder and synchronization circuit to produce synchronization between frame of ADSL DMT signal and sampling rate of PCM coder. The ADSL transmitter includes circuits for separate loading data and voice channels on "data carriers" and "voice carriers" of DMT line signal. The ADSL transmitter comprises a device for modulating of "voice carriers" with constellation table that minimize a noise in the voice channel. ATU-C comprises data interface port that connected to data switch of CO and voice interface port that connected to telephone switch of CO.

In another embodiment of the invention said ATU-C comprises PCM interface port connected to PCM telephone switch of CO and synchronization device to provide synchronization between PCM telephone switch and ADSL transceiver.

In another embodiment of the invention said ATU-R and said ATU-C include both circuits for provide a number of voice channels. Said ATU-R includes a number of PCM coders and a number of analog interface ports to provide a number of voice channels to customer. Said ATU-C includes multi-channel PCM interface port for example T1 interface port that connected to PCM telephone switch, and synchronization device to provide synchronization between PCM telephone switch and ADSL transceiver.

In another embodiment of the invention said ATU-R and said ATU-C include devices for dynamic allocation of data on" voice carriers" that are not used for telephone communication in the current time.

In another embodiment of the invention said ATU-R comprises a number of PCM coders, corresponding number of voice interface ports and PCM concentrator connected between said PCM coders and ADSL transmitter. An ADSL transmitter provides simultaneously several voice channels to CO. A number of voice channels of the ADSL transmitter less than number of said voice interface ports.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG.1 illustrates a block diagram a splitterless ADSL system of the prior art.

FIG.2 illustrates a block diagram of ATU transmitter reference model of the prior art.

FIG.3 is a bit-loading graph of ADSL of the prior art.

FIG.4 illustrates a block diagram of ADSL system in basic embodiment of the Invention.

FIG.5 illustrates a block diagram of ATU transmitter reference model in basic embodiment of the invention.

FIG.6 is a bit-loading graph of ADSL in basic embodiment of the invention.

FIG.7 is a timing diagram of ATU transmitter in basic embodiment of the invention.

FIG.8 is a constellation diagram of the QAM vector of the "voice carrier".

FIG.9 illustrates a block diagram of ATU in one embodiment of the invention.

FIG.10 illustrates a block diagram of ATU in one embodiment of the invention.

FIG.11 illustrates a block diagram of ATU in one embodiment of the invention.

FIG.12 illustrates a block diagram of ATU in one embodiment of the invention.

FIG.13 illustrates a block diagram of ATU in one embodiment of the invention.

DETAILED DESCRIPTION OF THE INVENTION

The present invention is directed to a method and apparatus for data and voice communication in an asymmetric digital subscriber line (ADSL) system. In accordance with the present invention, the disadvantages and problems associated with voice transmitting over ADSL system have been substantially reduced or eliminated. In particular, several high quality telephone channels may be inserted into ADSL system without necessary to change of existing standard.

In accordance with present invention a voice and data signals are simultaneously transmitted on different carriers of DMT ADSL line signal.

Fig.4 illustrates a communication system **201** in accordance with basic embodiment of the present invention. A subscriber premise **103** is coupled to central office (CO) **109** by twisted pair **107** of telephone cable. At the subscriber premise **103** said twisted pair **107** is connected to ATU-R **205**. A fax **121** and a telephone **123** are connected to an voice interface port **203** of the ATU-R **205** using internal telephone lines **117**. A personal computer (PC) **125** is connected to a digital interface port **204** of the ATU-R **205** by an Ethernet cable **124**. A Central Office (CO) **109** includes an ATU-C **211**, a data switch **135**, a telephone switch **137**, a data network **115**, and a telephone network **113**. A subscriber twisted pair **107** is coupled directly to the ATU-C **211** and carries data and voice by DMT line signals. A data passes from data interface port **209** of the ATU-C **211** to the data switch **135**. A voice passes from voice interface port **207** of the ATU-C **211** to the telephone switch **137**. The telephone switch **137** is coupled to a telephone network **113** and a data switch **135** is coupled to data network **115**. A communication system **201** not needs in expensive POTS-splitters and POTS Line Cards. In accordance with present invention voice signals are transmitted in digital form using part of capacity of ADSL link. A communication system **201** uses for transmitting voice and data signals different carriers ("voice carriers" and "data carriers") of ADSL DMT line signal.

Fig.5 is a block diagram of an ADSL Transceiver Unit (ATU) transmitter showing the functional blocks and interfaces in accordance with basic embodiment of the present invention. An information data **153** comes to a data interface port **151** that produce a sequence of ATM cells **152**. Said cells come to the coder block **155** that produces scrambling and Reed Solomon coding of ATM cells. After that a coded data stream **156** comes to interleaver **157** that mixes data bits to protect the information from impulse noise. An

interleaved data stream **158** passes to "data carriers" tone ordering block **263** that distributes this data on 126 "data carriers" of the DMT line signal. A "data carriers" constellation encoder and a gain-scaling block **265** calculate modulation parameters for each carrier in accordance with data that will be transmitted by said carrier. An Inverse Fourier Transformer block **163** transforms said modulation parameters of all carriers to digital samples of DMT signal and writes said digital samples into output buffer **165**. A digital to analog converter **167** transforms said digital samples to analog DMT line signal. A voice signal **251** comes to a voice interface port **253** that produces necessary amplifying and filtering of the said voice signal. A PCM coder **257** that connected to said voice interface port **253** transforms an analog voice signal in a 64 kbit/sec sequence of 8-bit PCM words. A samples rate of the PCM coder is 8 kHz. A PCM coder uses standard A – Law or μ-Law coding the same that is used in PCM telephone systems T1 or E1. A PCM words comes to a "voice carriers " tone-ordering block **259** that distributes 64 kb/sec PCM stream between two "voice carriers" of DMT signal. A "voice carriers" constellation encoder and gain-scaling block **261** transforms each 8-bit PCM word in one 8-bit QAM symbol and provides fixed 8-bit loading on each "voice carrier". A synchronization block **255** provides synchronization between samples rate of the PCM coder and frames of DMT line signal.

Fig.6 illustrates typical graph of ADSL downstream and upstream bit-loading in accordance with the present invention. Two of upstream carriers and two of downstream carriers are used for voice transmitting ("voice carriers"). "Voice carriers" are able to carry more then 8-bit per QAM symbol, but are modulated only with 8-bit QAM symbols. Each "voice QAM symbol" carries one 8-bit PCM word. Because samples rate of the PCM coder is 8 kHz and ADSL DMT frames follow with 4 kHz frequency there is necessary to use two "voice carriers" of upstream and two "voice carriers" of downstream for transmitting of one telephone channel. "Voice carriers" are not determinate before start of communication. In during initialization phase of communication an ADSL system measures a signal to noise ratio for each carrier and define number of bit that may be loaded on the carrier. Two of downstream and two of upstream curriers with highest signal to noise ratio those are able to carry more then 8-bit per symbol are reserved for voice transmission. As well known an ADSL standard is not define number of carriers those are used for data transmission. A part of carriers with low signal to noise ratio may be disabled by ADSL transmitter in during initialization process. In accordance with present invention a "voice carriers" perceives by

ADSL system as carriers those are not usable for data transmission. In this case ADSL system uses others DMT carriers for data transmission in accordance with G.992.2 standard.

Fig. 7 illustrates a time diagram of the said ADSL transmitter operation. Fig7.a shows frames of DMT line signals. Each frame of DMT has length of 250us and comprises one QAM symbol for each DMT carriers. Fig.7b shows a voice signal and its samples that are transformed by PCM coder into 8-bit PCM words. Fig.7c shows an allocation of said PCM words on first and second "voice carriers".

Fig.8 illustrates a "voice carriers" constellation encoder operation. Each 8-bit PCM word is transformed into 8-bit QAM vector. The constellation encoder calculates real and image components of the QAM vector using odd bits of PCM word for real component of said vector and even bits of PCM word for image component of said vector. More significant bits of PCM word are corresponding to more significant bits of real and image components of said QAM vector. A channel noise may produce errors only in less significant bits of the PCM words because low distance between corresponding QAM vectors. As result errors in low significant bits of PCM words produce only small additional noise in voice signal. In accordance with the present invention a voice signal may be transmitted with high quality over ADSL without any error correction coding and without significant delay.

It is clear that communication system Fig.5 may be extended for simultaneously transmitting of several voice channels.

Fig.9 is a block diagram of an ADSL Transceiver Unit (ATU) transmitter showing the functional blocks and interfaces in accordance with one embodiment of the present invention. An ADSL transmitter in this embodiment supports number of telephone channels. An information data **153** comes to a data interface port **151** that produce a sequence of ATM cells **152**. Said cells come to the coder block **155** that produces scrambling and Reed Solomon coding of ATM cells. After that a coded data stream **156** comes to interleaver **157** that mixes data bits to protect the information from impulse noise. An interleaved data stream **158** passes to "data carriers" tone ordering block **263** that distributes this data on "data carriers" of the DMT line signal. A "data carriers" constellation encoder and a gain-scaling block **265** calculate modulation parameters for each carrier in accordance with data that will be transmitted by said carrier. An Inverse Fourier Transformer block **163** transforms said modulation parameters of all carriers to digital samples of DMT signal and writes said digital samples into output buffer **165**. A digital to analog converter **167** transforms said digital

samples to analog DMT line signal. A number of voice signals **251** comes to a number of voice interface ports **253** those produce necessary amplifying and filtering of said voice signals. A number of PCM coders **257** are connected to correspond voice interface ports **253.** Each PCM coder transforms an analog voice signal in a 64-kbit/sec sequence of 8-bit PCM words. A samples rate of the PCM coder is 8 kHz. Each PCM coder uses standard A–Law or μ-Law coding the same that is used in PCM telephone systems T1 or E1. A PCM words comes to a "voice carriers " tone-ordering block **259** that distributes each 64 kb/sec PCM stream between two "voice carriers" of DMT signal. A Reed-Solomon coder (RSC) **258** calculates parity bytes for PCM words of all voice channels and puts said parity bytes on additional "voice carrier". A "voice carriers" constellation encoder and gain-scaling block **261** transforms each 8-bit PCM word in one 8-bit QAM symbol and provides fixed 8-bit loading on each "voice carrier". A synchronization block **255** provides synchronization between samples rate of each PCM coder and frames of DMT line signal.

Fig.10 illustrates a block diagram of an ATU transmitter in accordance with another embodiment of the present invention. An information data **153** comes to a data interface port **151** that produce a sequence of ATM cells **152**. Said cells come to the coder block **155** that produces scrambling and Reed Solomon coding of ATM cells. After that a coded data stream **156** comes to interleaver **157** that mixes data bits to protect the information from impulse noise. An interleaved data stream **158** passes to a carrier allocation block **301**. A carrier allocation block **301** is connected to processor **303** that analyzes telephone-signaling information that comes from voice interface blocks **253**. A carrier allocation block **301** distributes a data stream **158** between "data carriers" and "voice carriers" those are not busy in current time for transmitting a telephone channels. A communication system in accordance with said embodiment of the invention may carry more data because a part of telephone channels is not busy in during off-peak hours. A carrier allocation block **301** is connected to "data carriers" tone ordering block **263** and to "voice carriers" tone ordering block **259**. A number of voice signals **251** comes to a number of voice interface ports **253** those produce necessary amplifying and filtering of said voice signals. A number of PCM coders **257** are connected to correspond voice interface ports **253.** Each PCM coder transforms an analog voice signal in a 64-kbit/sec sequence of 8-bit PCM words. A samples rate of the PCM coder is 8 kHz. Each PCM coder uses standard A–Law or μ-Law coding the same that is used in PCM telephone systems T1 or E1. PCM words come to a carriers allocation block **301** and

after that to "voice carriers " tone-ordering block **259** that distributes each 64 kb/sec PCM stream of busy telephone channels between two "voice carriers" of DMT signal. A carriers allocation block **301** passes to "voice carriers " tone ordering block **259** only PCM signals from active PCM coders those provide telephone communication in the current time. Passive PCM coders those coupled to not used in current time telephone lines are not connected to "voice carriers " tone ordering block **259**. A Reed-Solomon coder (RSC) **258** calculates parity bytes for PCM words of active voice channels and puts said parity bytes on additional "voice carrier". A "voice carriers" constellation encoder and gain-scaling block **261** transforms each 8-bit PCM word in one 8-bit QAM symbol and provides fixed 8-bit loading on each "voice carrier". A synchronization block **255** provides synchronization between samples rate of each PCM coder and frames of DMT line signal.

Fig.11 illustrates a block diagram of an ATU-C transmitter in accordance with another embodiment of the present invention. Said ATU-C device is well adapted for operation together with electronic communication equipment of CO for example with PCM telephone switch (frame relay).

An information data **153** comes to a data interface port **151** that produce a sequence of ATM cells **152**. Said cells come to the coder block **155** that produces scrambling and Reed Solomon coding of ATM cells. After that a coded data stream **156** comes to interleaver **157** that mixes data bits to protect the information from impulse noise. An interleaved data stream **158** passes to "data carriers" tone ordering block **263** that distributes this data on "data carriers" of the DMT line signal. A "data carriers" constellation encoder and a gain-scaling block **265** calculate modulation parameters for each carrier in accordance with data that will be transmitted by said carrier. An Inverse Fourier Transformer block **163** transforms said modulation parameters of all carriers to digital samples of DMT signal and writes said digital samples into output buffer **165**. A digital to analog converter **167** transforms said digital samples to analog DMT line signal.

A telephone signal comes to the ATU-C from said frame relay directly in the digital form. A 64 kbit/s PCM stream **281** comes to an input of a PCM interface block **283** that provides necessary synchronization between frames of DMT signal and PCM 8-bit words. Said 64kb/s data stream comes to a "voice carriers " tone-ordering block **259** that distributes this 64 kb/sec PCM stream between two "voice carriers" of DMT signal. A "voice carriers" constellation encoder and gain-scaling block **261** transforms each 8-bit PCM word in one 8-

bit QAM symbol and provides fixed 8-bit loading on each "voice carrier". A main 8-kHz clock **285** of the frame relay comes to the synchronization block **255** and to PCM interface block **283** to provide a synchronism between a frame relay and an ADSL system.

Fig.12 illustrates a block diagram of an ATU-C transmitter in accordance with another embodiment of the present invention. Said ATU-C device is well adapted for operation together with electronic communication equipment of CO for example with PCM telephone switch (frame relay) that has T1 interface. An information data **153** comes to a data interface port **151** that produce a sequence of ATM cells **152**. Said cells come to the coder block **155** that produces scrambling and Reed Solomon coding of ATM cells. After that a coded data stream **156** comes to interleaver **157** that mix data bits to protect the information from impulse noise. An interleaved data stream **158** passes to "data carriers" tone ordering block **263** that distributes this data on "data carriers" of the DMT line signal. A "data carriers" constellation encoder and a gain-scaling block **265** calculate modulation parameters for each carrier in accordance with data that will be transmitted by said carrier. An Inverse Fourier Transformer block **163** transforms said modulation parameters of all carriers to digital samples of DMT signal and writes said digital samples into output buffer **165**. A digital to analog converter **167** transforms said digital samples to analog DMT line signal.

A number of voice (telephone) channels come from PCM telephone switch directly in the digital format of the T1 system. A T1 data stream **271** and T1 system clock **273** come to T1 interface block **277** that distribute T1 data stream into number of channels data streams **279**. Each 64 kb/s channels data stream **279** comprises sequence of 8-bit PCM words of a corresponding telephone channel. Said 64kb/s data streams come to a "voice carriers " tone-ordering block **259** that distributes each 64 kb/sec PCM stream between two "voice carriers". of DMT signal. A "voice carriers" constellation encoder and gain-scaling block **261** transforms each 8-bit PCM word in one 8-bit QAM symbol and provides fixed 8-bit loading on each "voice carrier". A synchronization block **255** provides synchronization between T1 system clock and frames of DMT line signal.

Fig.13 illustrates a block diagram of an ATU-R transmitter in accordance with another embodiment of the present invention. Said ATU-R device provides additionally function of a home telephone commutation. An information data **153** comes to a data interface port **151** that produce a sequence of ATM cells **152**. Said cells come to the coder block **155** that produces scrambling and Reed Solomon coding of ATM cells. After that a

coded data stream **156** comes to interleaver **157** that mix data bits to protect the information from impulse noise. An interleaved data stream **158** passes to "data carriers" tone ordering block **263** that distributes this data on "data carriers" of the DMT line signal. A "data carriers" constellation encoder and a gain-scaling block **265** calculate modulation parameters for each carrier in accordance with data that will be transmitted by said carrier. An Inverse Fourier Transformer block **163** transforms said modulation parameters of all carriers to digital samples of DMT signal and writes said digital samples into output buffer **165**. A digital to analog converter **167** transforms said digital samples to analog DMT line signal.

A number of voice signals **251** comes to a number of voice interface ports **253** those produce necessary amplifying and filtering of said voice signals. A number of PCM coders **257** are connected to correspond voice interface ports **253.** Each PCM coder **257** transforms an analog voice signal in a 64-kbit/sec sequence of 8-bit PCM words. A samples rate of the PCM coder is 8 kHz. Each PCM coder uses standard A–Law or μ-Law coding the same that is used in PCM telephone systems T1 or E1. Each PCM coder **257** is connected to a PCM concentrator **309**. An output of the concentrator **309** is connected to a "voice carriers" tone-ordering block **259** that distributes each 64 kb/sec PCM stream between two "voice carriers" of DMT signal. Said PCM concentrator **309** is able to support several telephone channels simultaneously. A number of said telephone channels less than number of PCM coders. For example 8 PCM coders may be connected to said PCM concentrator **309** that uses only 4 voice carriers to provide two telephone channels simultaneously.

It is understand that present invention may be realized not only in ADSL system but so in another communication systems that use DMT line signals for example in DMT-VDSL system.

Although the present invention has been described with several embodiments, a myriad of changes, variations, alterations, and modifications may be suggested to one skilled in the art, and it is intended that the present invention encompass such changes, variations, alterations, and modifications as fall within the spirit and scope of appended claims.

CLAIMS

The claimed invention is as follows:

1. A method for transmitting of voice and data over subscriber twisted pair, comprising: using of DMT line signal that comprises many carriers: each carrier is modulated by sequence of QAM symbols and transmits number of bit of information per QAM symbol in accordance with signal to noise ratio,
using different carriers of DMT line signal for simultaneously transmitting data and voice,
assigning for transmitting of voice a number of said carriers that may transmit equal or more than n bit per QAM symbol for transmitting of voice ,
assigning other carriers of DMT line signal for transmitting of data,
converting of voice in sequence of n-bit digital words by PCM coding ,
transmitting of said QAM symbols in synchronization with said sequence of PCM words,
transmitting of each n-bit PCM word by one n-bit QAM symbol on carriers assigned for voice transmission: each n-bit PCM word is converted in n-bit QAM symbol in such way that probability of error is minimal for more significant bits of said PCM word.

2. A method of claim 1 further comprising:
transmitting several voice signals on several carriers of DMT line signal assigned for voice transmitting,
using a number of said carriers assigned for voice transmission in during time intervals those are not used for telephone communication for data transmission ,

3. A method of claim 1 further comprising:
using of telephone signaling for decision which telephone channel is active and needs in transmitting a voice ,
transmitting voice signals of active telephone channels on several carriers of DMT line signal assigned for voice transmitting.

4. A xDSL transmitter for transmitting of voice and data over subscriber twisted pair comprising:
A digital to analog converter connected to subscriber twisted pair and coupled to

IFFT device that produce DMT signal,

A first constellation encoder connected to IFFT device to modulate number of carriers of DMT signal assigned for voice transmission by n-bit QAM symbols,

A second constellation encoder connected to IFFT device to modulate other carriers of DMT signal assigned to data transmission,

A first tone-ordering block connected to the first constellation encoder and to PCM coder to assignee number of carriers of DMT signal for voice transmitting,

A second tone-ordering block coupled to data interface port to assignee other carriers of DMT signal for data transmission,

A data interface port connected to external information source,

A PCM coder connected to voice interface port to transform a voice into sequence of n-bit PCM words,

A voice interface port connected to external voice source,

A synchronization device connected to the PCM coder and to IFFT device to provide synchronization between said sequence of n-bit PCM words and frames of DMT signal.

5. A xDSL transmitter of claim 4 further comprising:

A number of PCM coders connected to a number of voice interface ports: each PCM coder transforms a corresponding voice signal into sequence of n-bit PCM words,

A number of voice interface ports connected to number of external voice source to provide number of telephone channels,

A synchronization device connected to each PCM coder and to IFFT device to provide synchronization between sample rate of each PCM coder and frames of DMT signal.

6. A xDSL transmitter of claim 5 further comprising:

A carriers allocation block connected to first tone ordering block, to second tone ordering block and to a signaling processor to provide dynamic allocation of data on carriers of DMT signal assigned for voice transmitting those are not busy in the current time by telephone signals,

A signaling processor connected to telephone signaling circuit of each voice interface port to decide which telephone channel is busy.

7. A xDSL transmitter of claim 4 comprising:

A digital to analog converter connected to subscriber twisted pair and coupled to IFFT device that produce DMT signal,

A first constellation encoder connected to IFFT device to modulate number of carriers of DMT signal assigned for voice transmission by n-bit QAM symbols,

A second constellation encoder connected to IFFT device to modulate other carriers of DMT signal assigned to data transmission,

A first tone-ordering block connected to the first constellation encoder and to PCM interface port to assignee number of carriers of DMT signal for voice transmitting,

A second tone-ordering block coupled to data interface port to assignee other carriers of DMT signal for data transmission,

A data interface port connected to external information source,

A PCM interface port connected to external PCM telephone equipment,

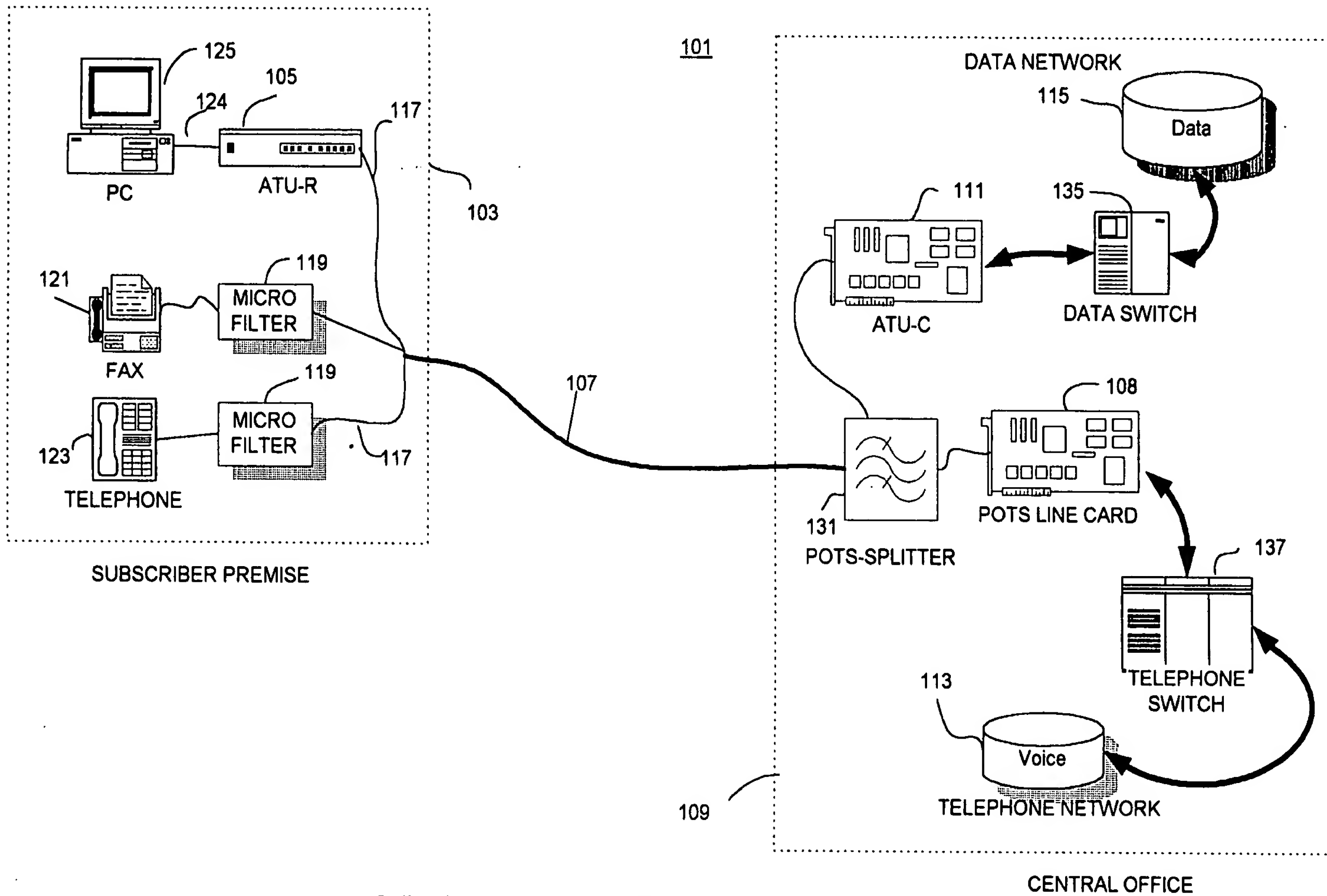
A synchronization device connected to the said external PCM equipment and to IFFT device to provide synchronization between frames of said PCM equipment and frames of DMT signal.

8. A xDSL transmitter of claim 5 further comprising:

A PCM concentrator connected to a number of said PCM coders and to first tone ordering block; a number of PCM coders is bigger than number of voice channels supported by xDSL transmitter.

INVENTOR
ROMAN VITENBERG

16.04.2000



Splitterless Acymmetric Digital Subscriber Line

FIG.1

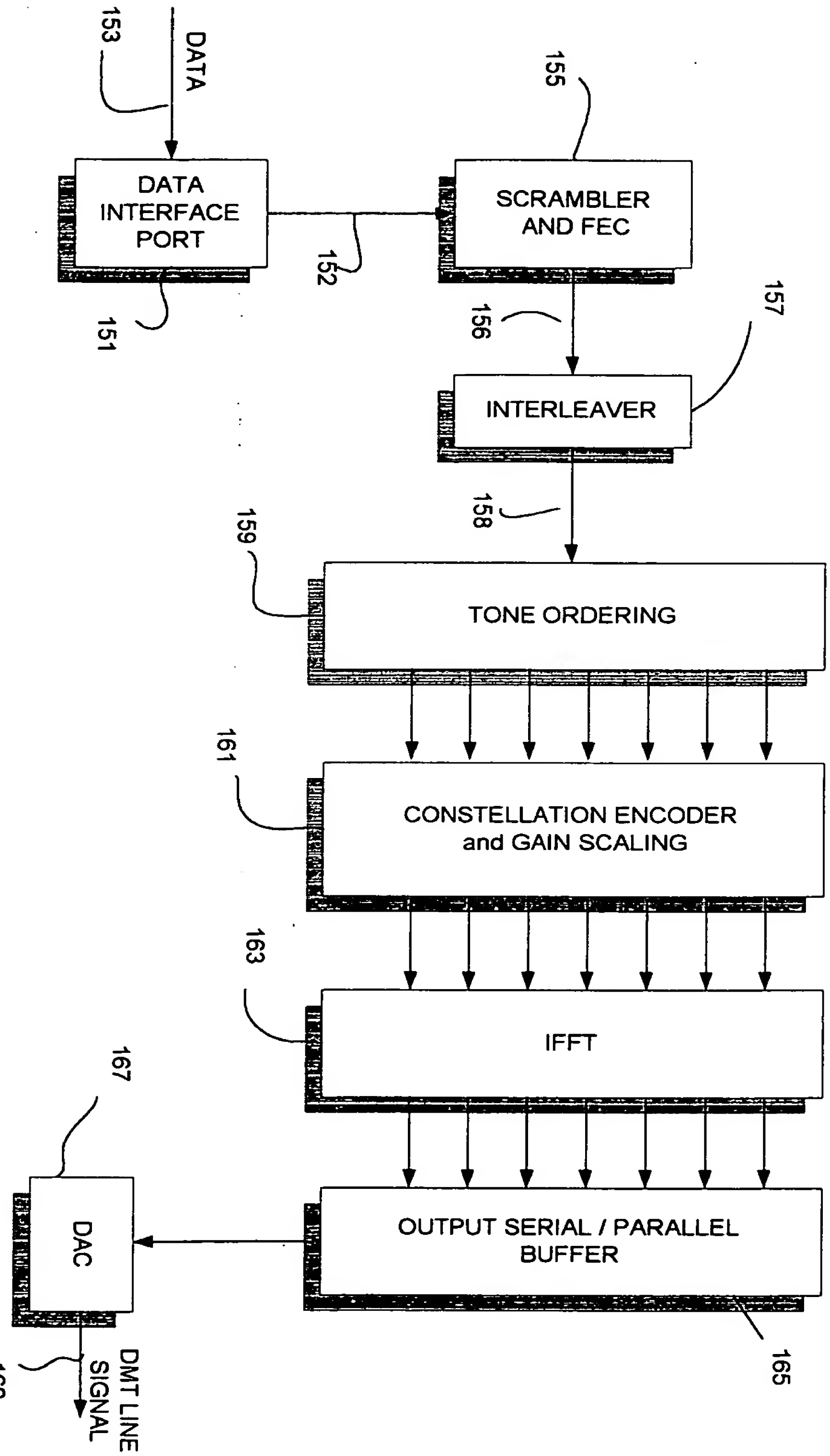


FIG.2 ATU REFERENCE MODEL

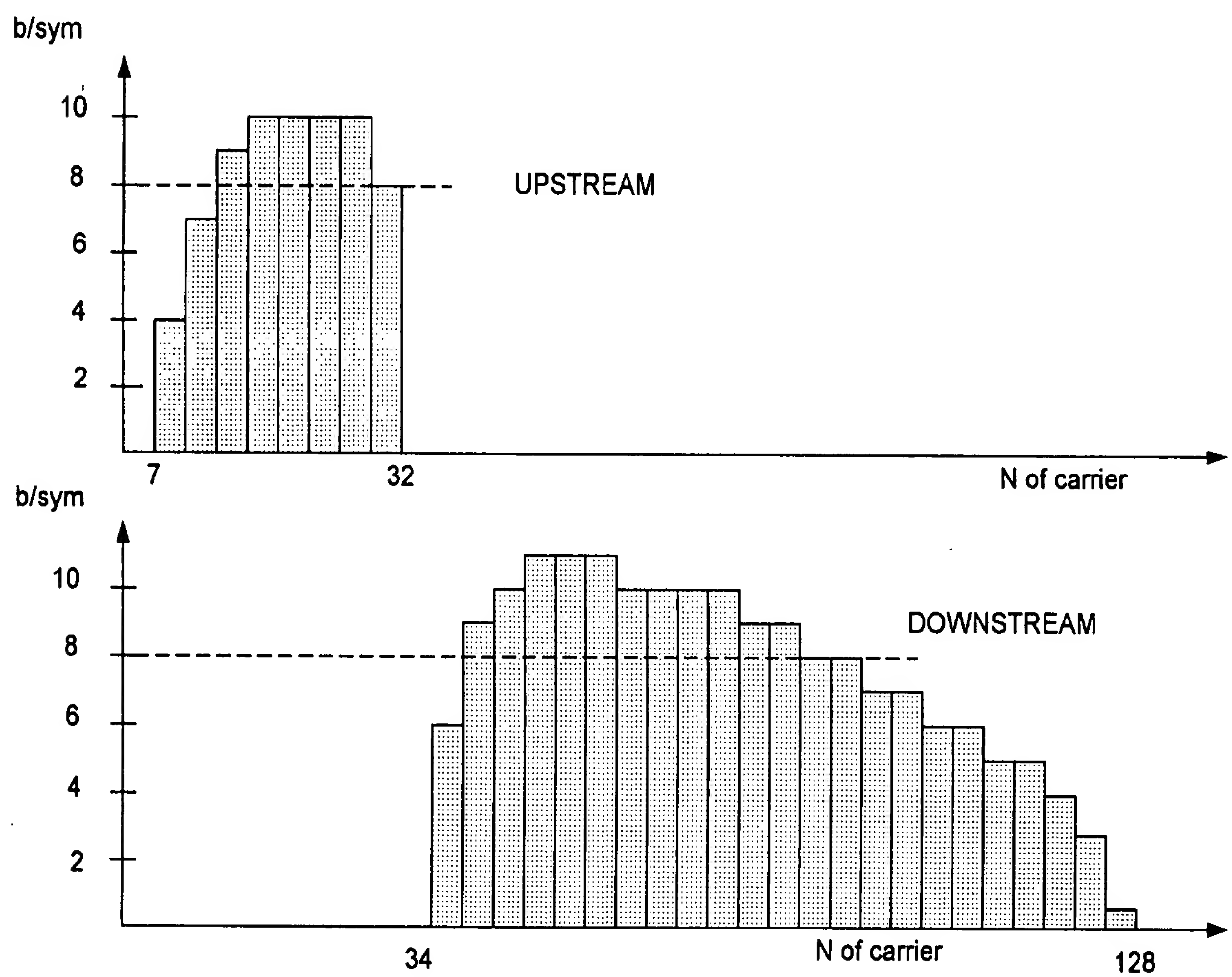

b/sym
10
8
6
4
2
7
32
UPSTREAM
N of carrier
b/sym
10
8
6
4
2
34
DOWNSTREAM
N of carrier
128

FIG.3

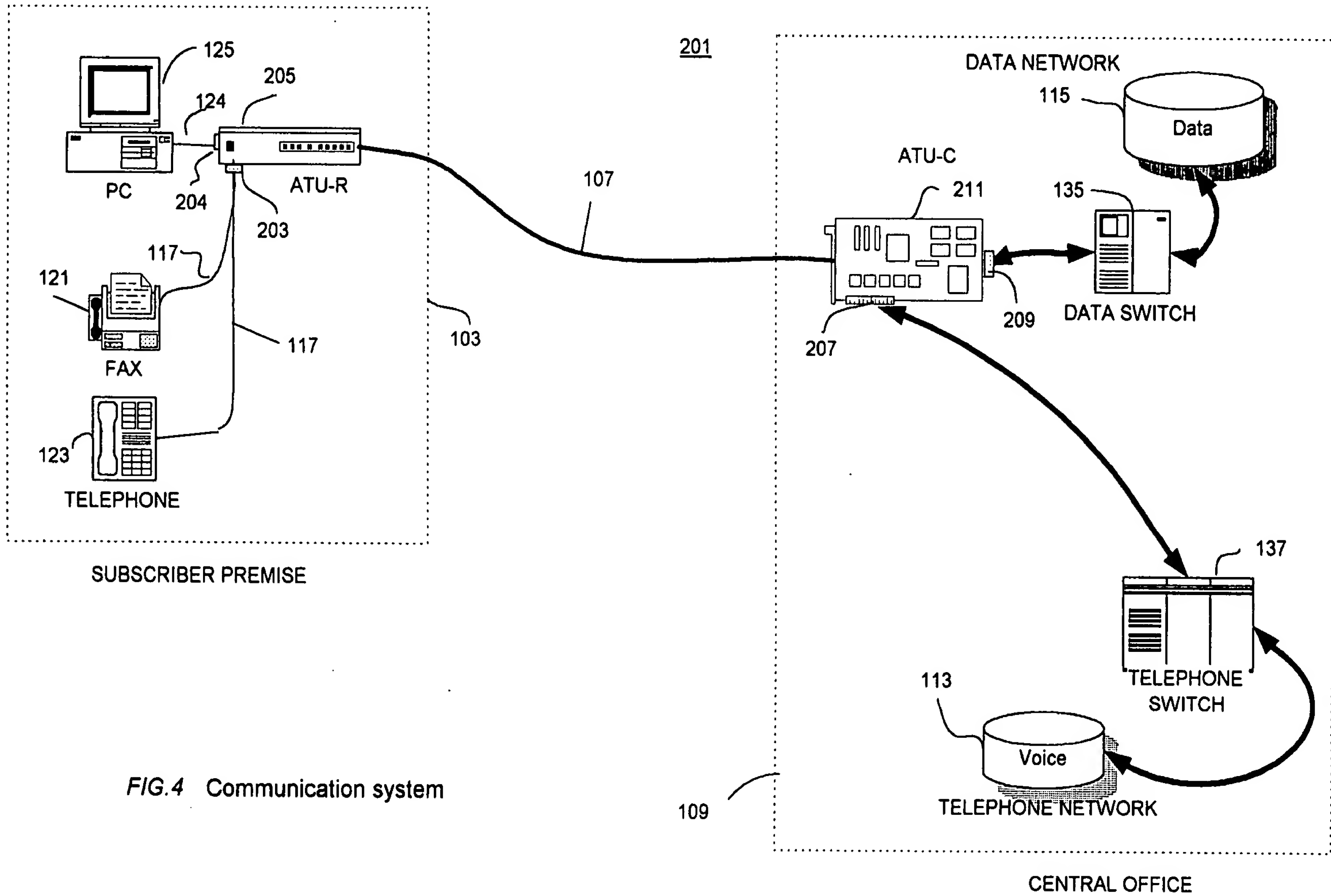


FIG.4 Communication system

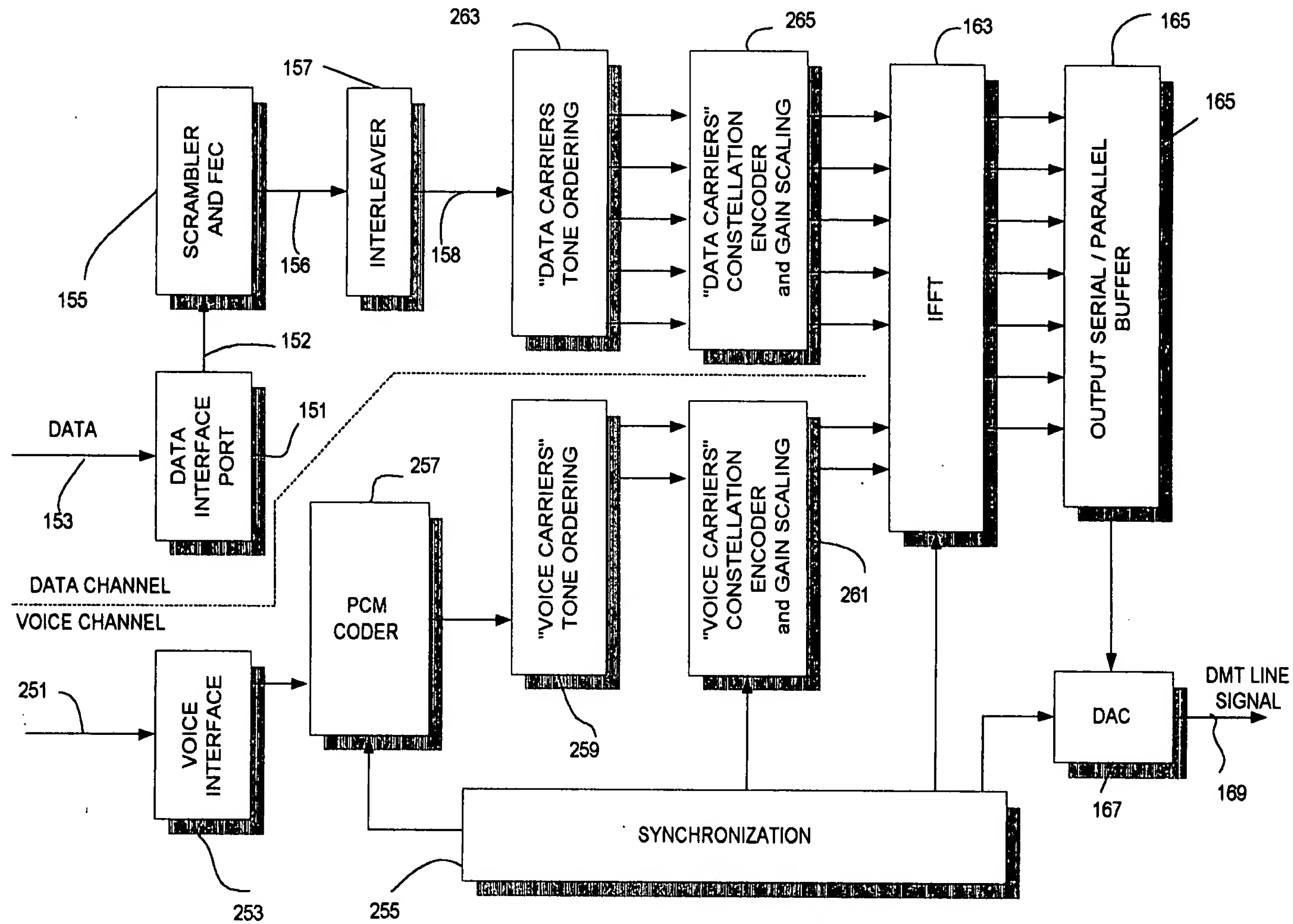


FIG.5 ATU REFERENCE MODEL

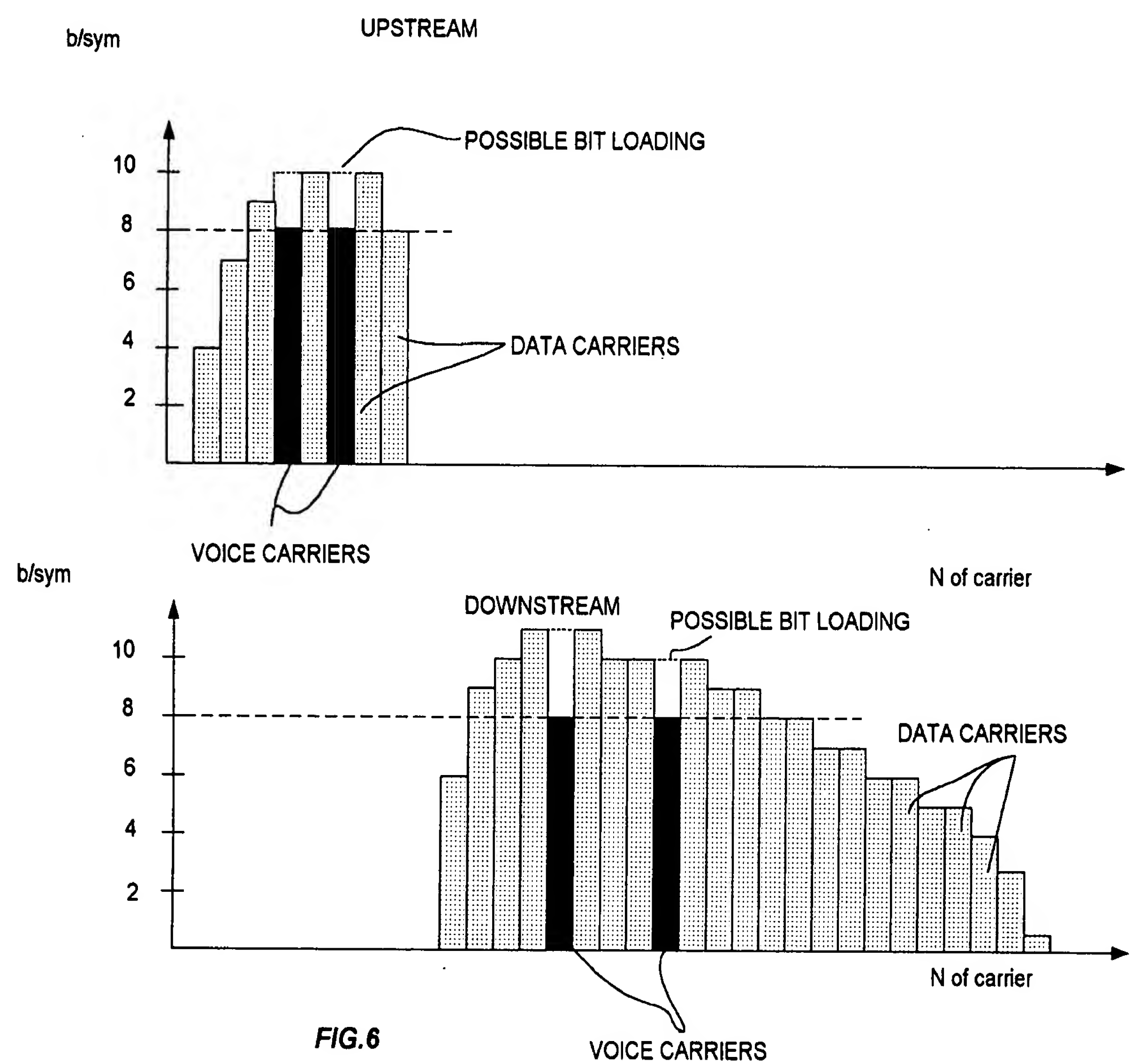
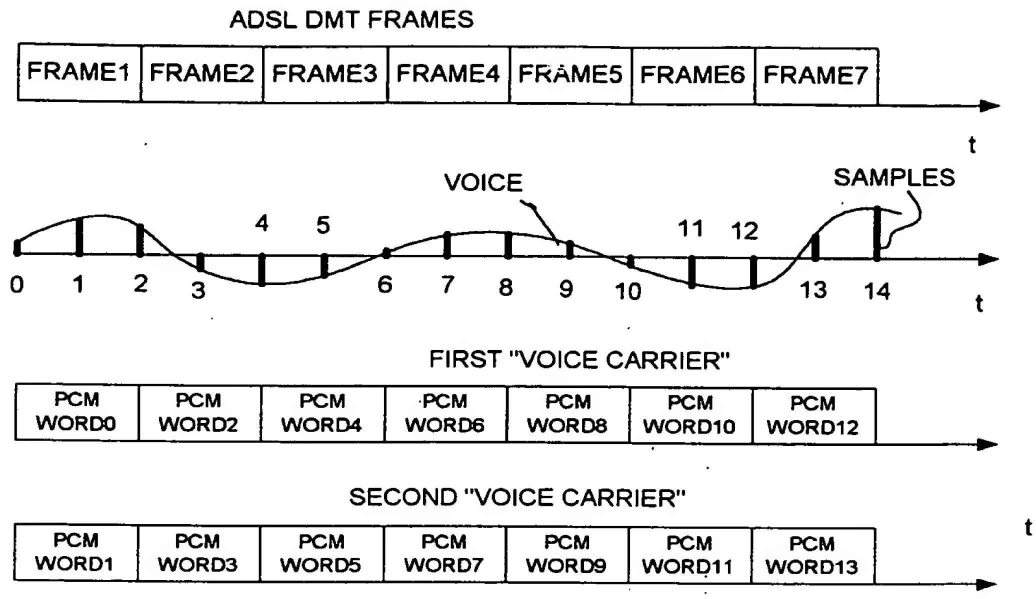
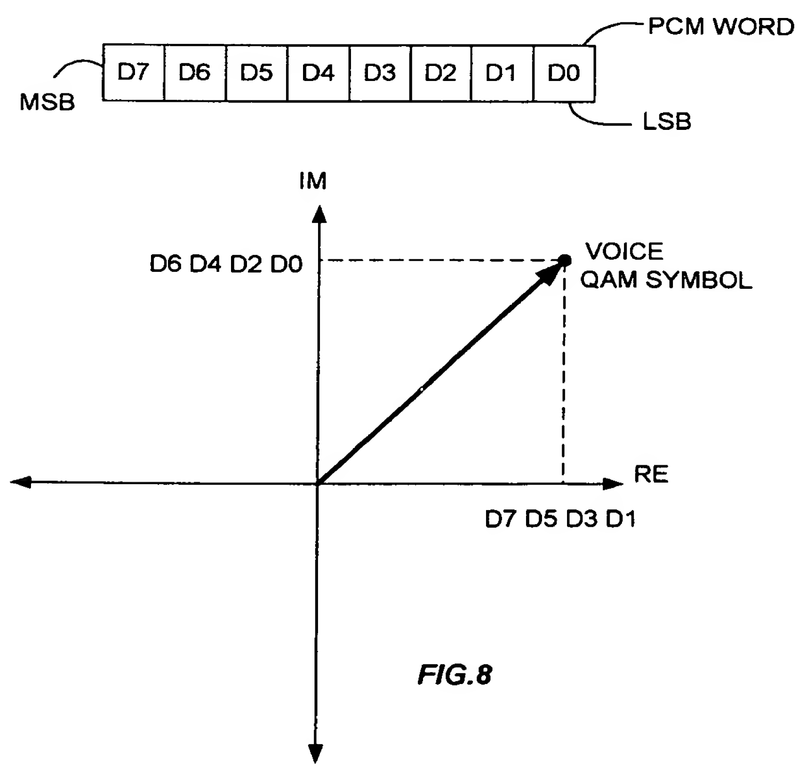

b/sym
UPSTREAM
10
8
6
4
2
POSSIBLE BIT LOADING
DATA CARRIERS
VOICE CARRIERS
N of carrier
b/sym
DOWNSTREAM
POSSIBLE BIT LOADING
10
8
6
4
2
DATA CARRIERS
N of carrier
VOICE CARRIERS

FIG.6



ADSL DMT FRAMES
FRAME1
FRAME2
FRAME3
FRAME4
FRAME5
FRAME6
FRAME7
t
VOICE
SAMPLES
0
1
2
3
4
5
6
7
8
9
10
11
12
13
14
t
FIRST "VOICE CARRIER"
PCM WORD0
PCM WORD2
PCM WORD4
PCM WORD6
PCM WORD8
PCM WORD10
PCM WORD12
SECOND "VOICE CARRIER"
PCM WORD1
PCM WORD3
PCM WORD5
PCM WORD7
PCM WORD9
PCM WORD11
PCM WORD13
t

FIG.7



PCM WORD
D7
D6
D5
D4
D3
D2
D1
D0
MSB
LSB
IM
D6 D4 D2 D0
VOICE QAM SYMBOL
RE
D7 D5 D3 D1

FIG.8

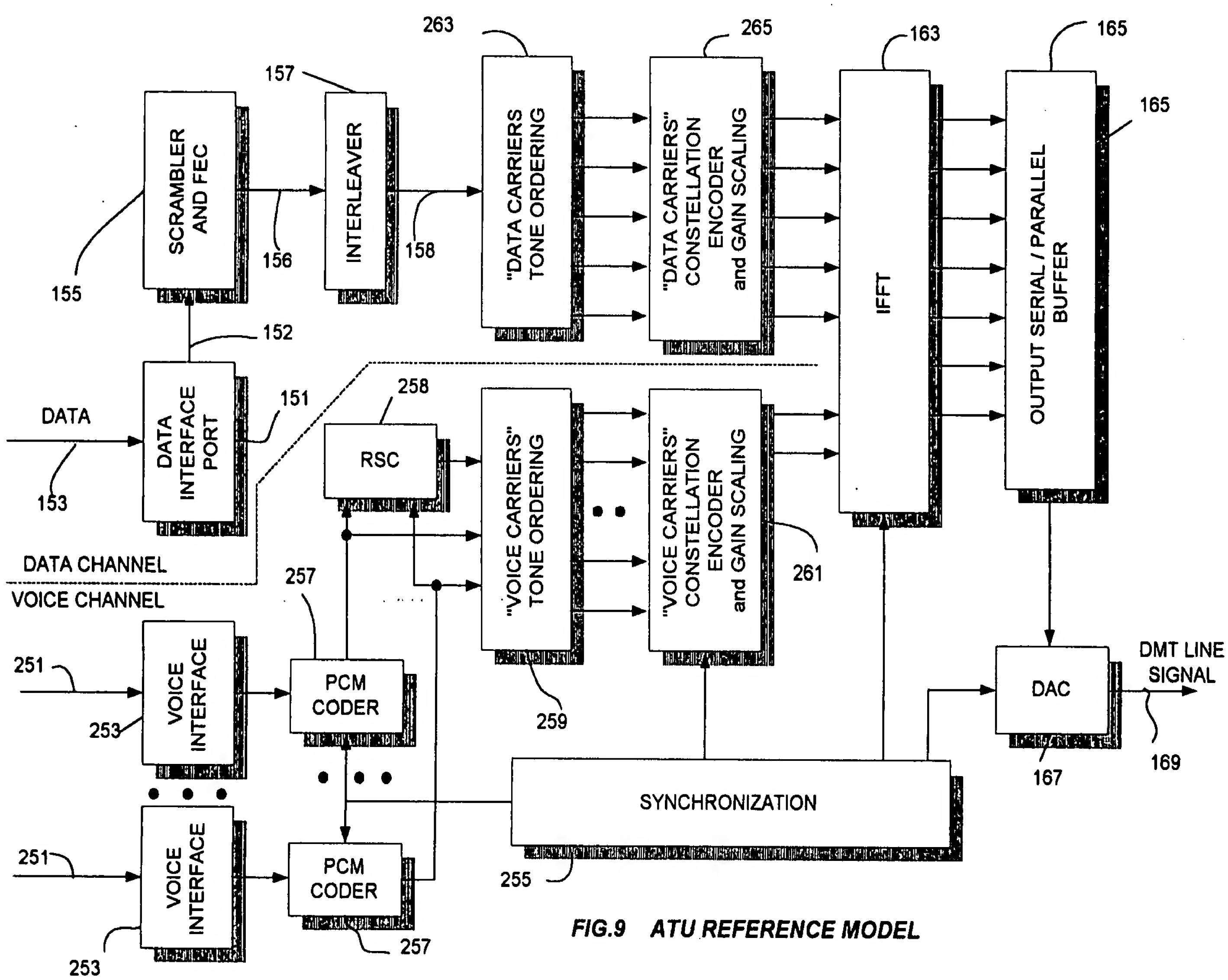


FIG.9 ATU REFERENCE MODEL

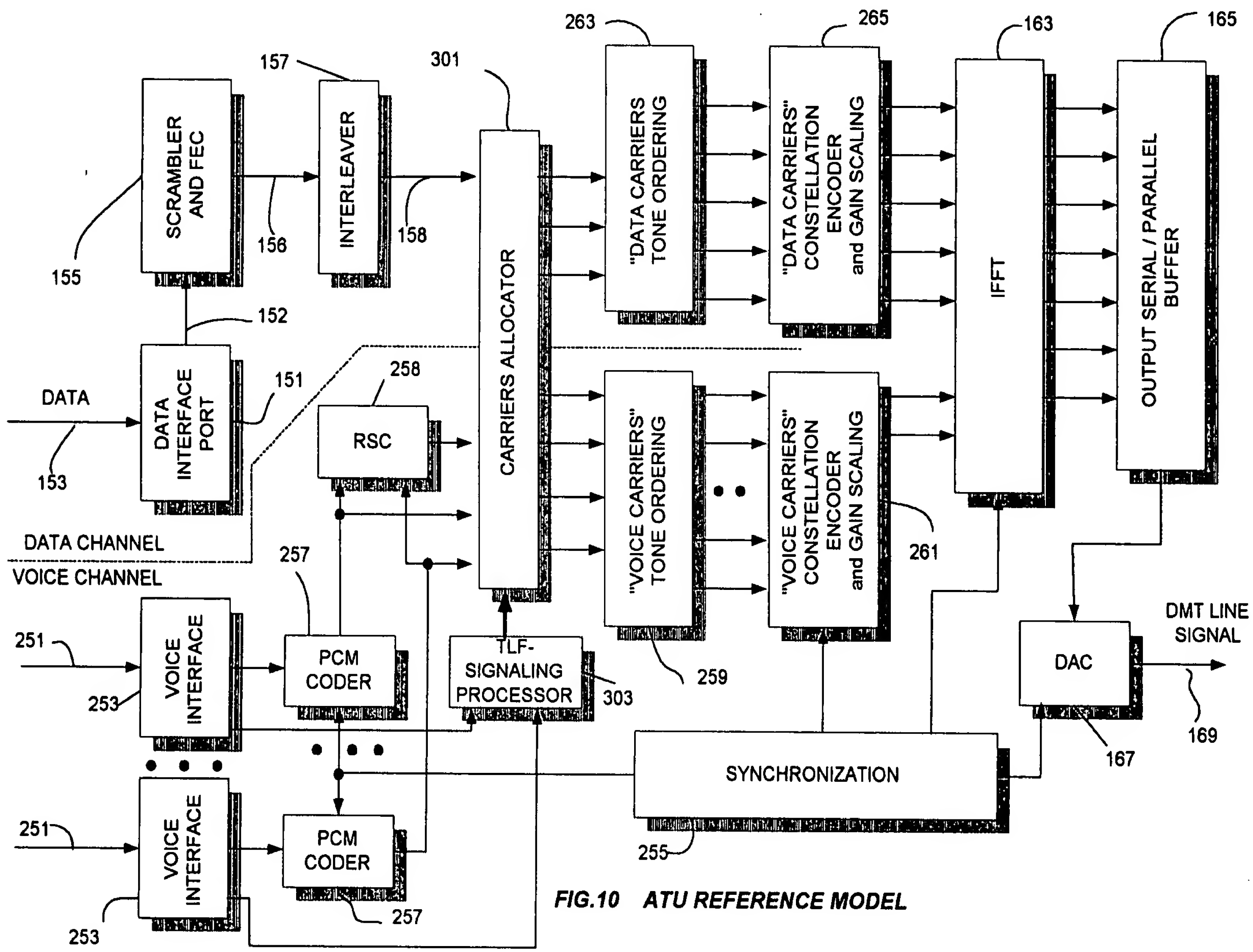


FIG.10 ATU REFERENCE MODEL

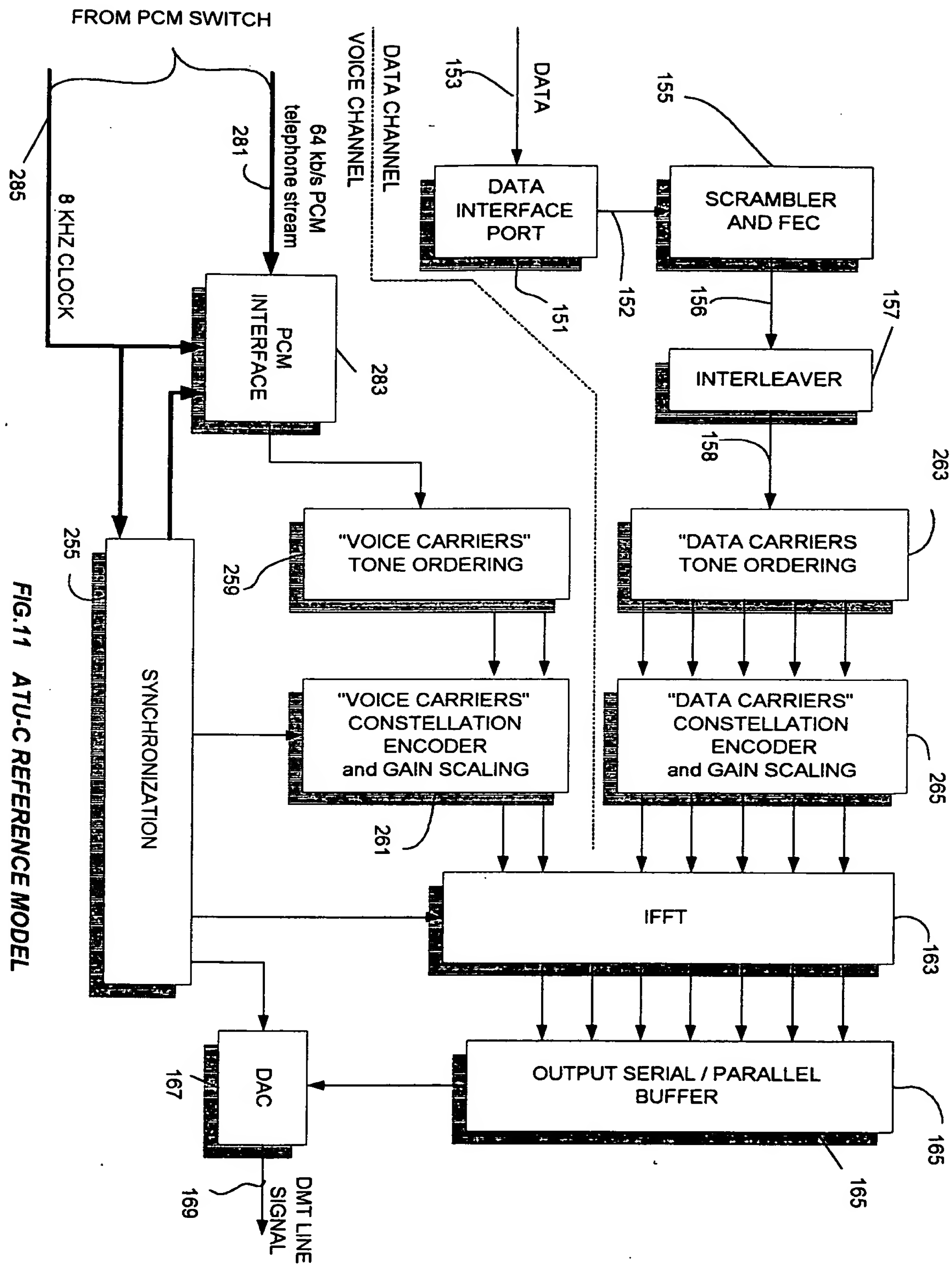


FIG.11 ATU-C REFERENCE MODEL

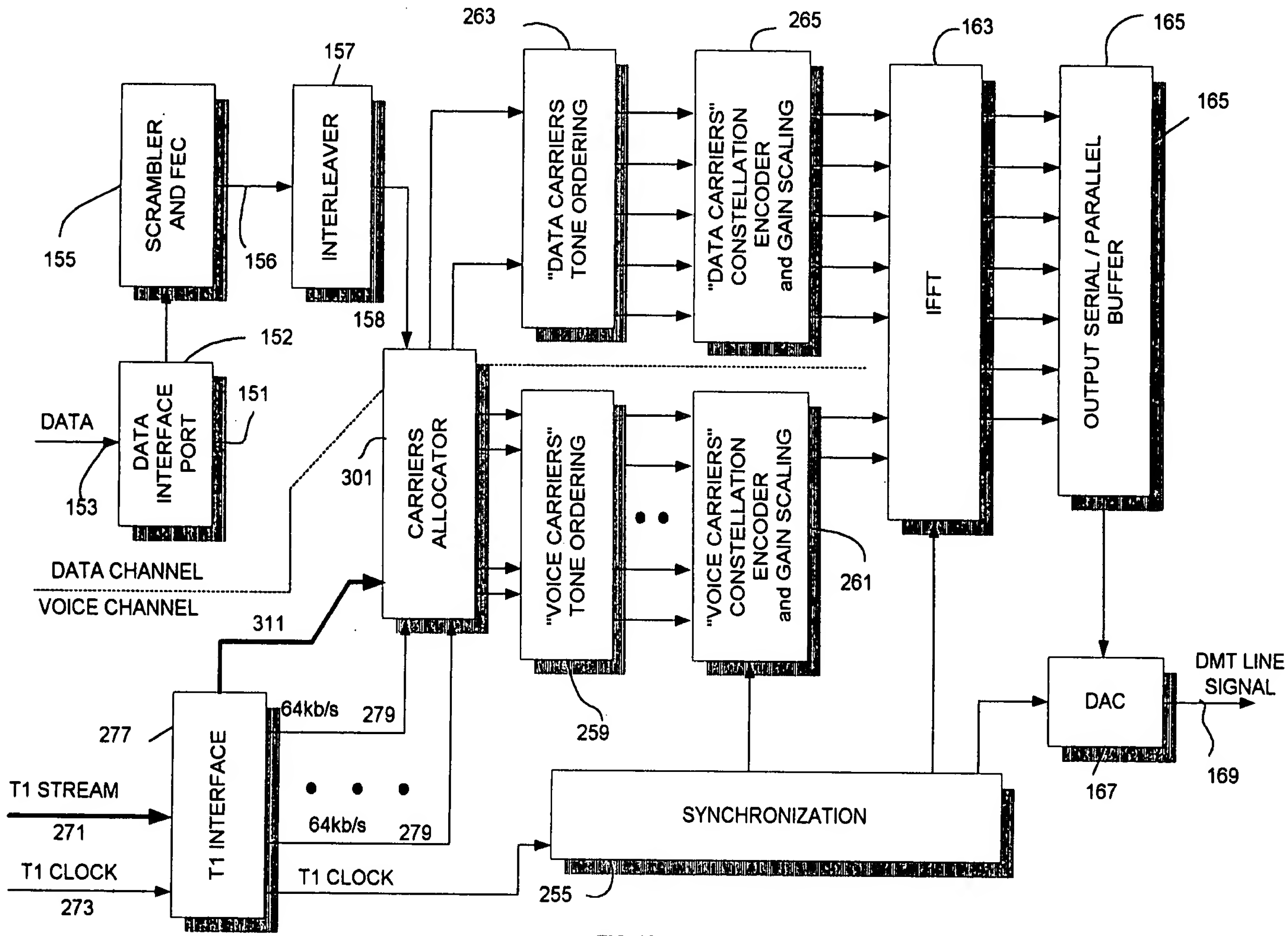


FIG.12 ATU-C REFERENCE MODEL

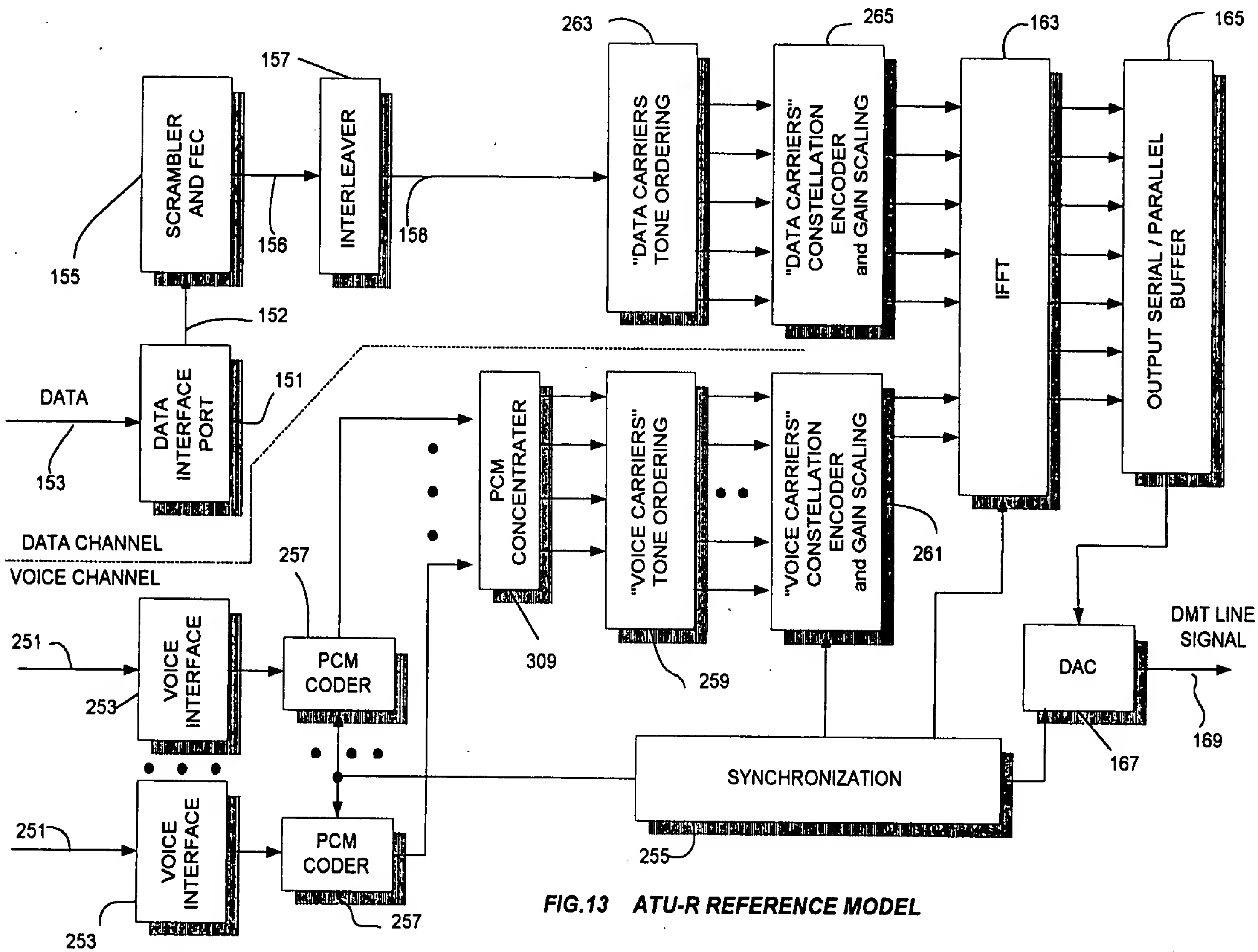


FIG.13 ATU-R REFERENCE MODEL